3. Is the analyst's information correct that FFT techniques produce more accurate averages than any others? Why or why not?

Problem 5.32: Echoes

Echoes not only occur in canyons, but also in auditoriums and telephone circuits. In one situation where the echoed signal has been sampled, the input signal $x(n)$ emerges as $x(n) + a_1 x(n - n_1) + a_2 x(n - n_2)$.

1. Find the difference equation of the system that models the production of echoes.
2. To simulate this echo system, ELEC 241 students are asked to write the most efficient (quickest) program that has the same input-output relationship. Suppose the duration of $x(n)$ is 1,000 and that
$$a_1 = \frac{1}{2},\ n_1 = 10,\ a_2 = \frac{1}{5},\ and\, n_2 = 25$$
Half the class votes to just program the difference equation while the other half votes to program a frequency domain approach that exploits the speed of the FFT. Because of the undecided vote, you must break the tie. Which approach is more efficient and why?
3. Find the transfer function and difference equation of the system that suppresses the echoes. In other words, with the echoed signal as the input, what system's output is the signal $x(n)$?

Problem 5.33: Digital Filtering of Analog Signals RU Electronics wants to develop a filter that would be used in analog applications, but that is implemented digitally. The filter is to operate on signals that have a 10 kHz bandwidth, and will serve as a lowpass filter.

1. What is the block diagram for your filter implementation? Explicitly denote which components are analog, which are digital (a computer performs the task), and which interface between analog and digital worlds.
2. What sampling rate must be used and how many bits must be used in the A/D converter for the acquired signal's signal-to-noise ratio to be at least 60 dB? For this calculation, assume the signal is a sinusoid.
3. If the filter is a length-128 FIR filter (the duration of the filter's unit-sample response equals 128), should it be implemented in the time or frequency domain?
4. Assuming $H(e^{j2\pi f})$ is the transfer function of the digital filter, what is the transfer function of your system?

Problem 5.34: Signal Compression

Because of the slowness of the Internet, lossy signal compression becomes important if you want signals to be received quickly. An enterprising 241 student has proposed a scheme based on frequency-domain processing. First of all, he would section the signal into length-N blocks, and compute its N-point DFT. He then would discard (zero the spectrum) at **half** of the frequencies, quantize them to b-bits, and send these over the network. The receiver would assemble the transmitted spectrum and compute the inverse DFT, thus reconstituting an N-point block.